What is claimed is:

1. A method comprising:

generating audio packets representing an input audio signal;

communicating the audio packets over a network;

generating an output audio signal from the communicated audio packets;

generating an input envelope waveform and an output envelope waveform from the input audio signal and the output audio signal, respectively; and

comparing the envelope waveforms.

2. The method of claim 1, wherein comparing the envelope waveforms includes subtracting the output envelope waveform from the input envelope waveform.

3. The method of claim 1, wherein comparing the envelope waveforms includes determining a transmission quality including at least one of data loss and latency.

4. The method of claim 1, wherein communicating the audio packets includes communicating the audio packets over the Internet.

5. The method of claim 1, wherein communicating the audio packets includes communicating the audio packets between telephony-enabled computers.

6. The method of claim 1, wherein generating the audio packets includes generating the audio packets from an Internet telephone.

7. The method of claim 1, wherein:

generating the audio packets includes digitizing the input audio signal and compressing the digitized input audio signal using an input coder/decoder (codec) having a first buffer length,

generating the output audio signal includes generating the output audio signal using an output coder/decoder (codec) having a second buffer length, and

generating the envelope waveforms includes generating the envelope waveforms at a resolution that is a function of the first buffer length and the second buffer length.

8. A method comprising:

capturing an input audio signal and an output audio signal associated with a network based telephony communication;

generating an input envelope waveform and an output envelope waveform from the input audio signal and the output audio signal, respectively; and

subtracting the output envelope waveform from the input envelope waveform to produce a summary envelope waveform.

9. The method of claim 8, wherein generating the input and output envelope waveforms includes removing a bias.

10. The method of claim 8, wherein generating the input and output envelope waveforms includes normalizing the captured input and output audio signals.

11. The method of claim 8, wherein capturing the input and output audio signals includes triggering the capture using a trigger signal embedded within the input audio signal.

12. The method of claim 8, wherein generating the input and output envelope waveforms includes aligning the captured input and output audio signals.

13 The method of claim 8, wherein the output audio signal comprises an analog signal generated from an audio data stream of digital packets communicated over a packet-based network, and further wherein the digital data stream is generated from the input audio signal.

14. The method of claim 13, wherein generating the input and output envelope waveforms includes generating the envelope waveforms at a resolution that is a function of a buffer length of coder/decoders (codecs) used in generating the audio data stream and the output audio signal.

15. An article comprising a computer-readable medium having computer-executable instructions stored thereon for causing a computer to:

capture an input audio signal and an output audio signal associated with a network based telephony communication;

generate an input envelope waveform and an output envelope waveform from the input audio signal and the output audio signal, respectively; and

subtract the output envelope waveform from the input envelope waveform to produce a summary envelope waveform.

16. The article of claim 15, wherein the computer-executable instructions cause the computer to generate the input and output envelope waveforms by removing any amplitude bias in the captured signals, normalizing the captured signals, and aligning the captured signals using a trigger signal embedded within the input audio signal.

17. The article of claim 15, wherein the output audio signal is an analog signal generated from an audio data stream of digital packets communicated over a packet-based network, and further wherein the digital data stream is generated from the input audio signal.

18. The article of claim 17, wherein the computer-executable instructions cause the computer to generate the envelope waveforms at a resolution that is a function of a buffer length of coder/decoders (codecs) used in generating the audio data stream and the output audio signal.

19. A system comprising:

a transmit device to convert an input audio signal to data packets;

a receive device communicatively coupled to the transmit device via a packet switched network, wherein

the receive device receives the data stream and converts the data stream to an output audio signal; and

an audio analyzer coupled to the transmit device and the receive device, wherein the audio analyzer captures the input audio signal and the output audio signal, and further wherein the audio analyzer generates a data loss summary envelope from the input audio signal and the output audio signal.

20. The system of claim 19, wherein the transmit device includes a coder/decoder (codec) to convert the input audio signal to digital data and the receive device includes a coder/decoder (codec) to convert the digital data stream to the output audio signal, and further wherein the summary envelope has a resolution that is as a function of a buffer length of the codec of the transmit device and a buffer length for the codec of the receive device.

21. The system of claim 20, wherein the codecs have equal buffer lengths and the resolution of the envelope waveforms is approximately 25% of the codec buffer length.

22. The system of claim 20, wherein the codecs are G.723 codecs and the transmit device communicates the data stream using the H.323 protocol, and further wherein the buffer length is approximately 30ms and the resolution of the envelope waveforms is approximately 7.5ms.

23. The system of claim 19, wherein the network is a global computer network

24. The system of claim 19, wherein the transmitting device or the receiving device comprises an telephony-enabled computer.

25. The system of claim 19, wherein the transmitting device or the receiving device comprises an Internet telephone.

26. The system of claim 19, wherein the audio analyzer further includes means for subtracting the output audio signal from the input audio signal to generate the summary data loss envelope.

27. The system of claim 19, wherein the audio analyzer includes a graphical user interface that displays in real-time the summary envelope waveform and transmission qualities within the audio test system including latency.

28. The system of claim 19, wherein the audio analyzer includes a multi-channel dynamic signal analyzer for sampling the input audio signal and the output audio signal.

29. The system of claim 19 and further including an audio generator to generate the input audio signal from a stored audio file.

30. The system of claim 19, wherein the input audio signal includes a trigger signal having a low-frequency, high amplitude pulse.